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(54) Automated speech alignment for image synthesis

(57) In a computerized method, speech signals are analyzed using statistical trajectory modeling to produce time aligned acoustic-phonetic units. There is one acoustic-phonetic unit for each portion of the speech signal determined to be phonetically distinct. The acoustic-phonetic units are translated to corresponding time aligned image units representative of the acoustic-phonetic units. An image including the time aligned image units is displayed. The display of the time aligned image units is synchronized to a replaying of the digitized natural speech signal.

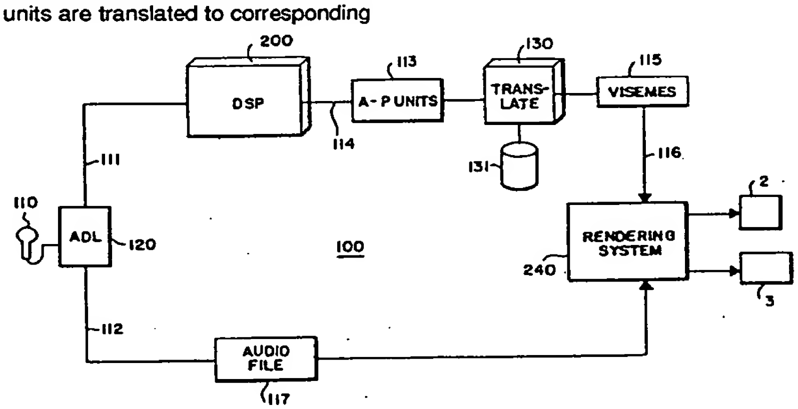

FIG. 1

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Description

FIELD OF THE INVENTION

The present invention relates generally to audio-visual signal processing, and more particularly to aligning speech signals with synthetically generated facial images.

BACKGROUND OF THE INVENTION

For some computer applications, it is desired to dynamically time-align an animated image with audio signals. For example, most modern computers are commonly equipped with a "sound-card." The sound card can process and reproduce audio signals such as music and speech. In the case of speech, the computer can also dynamically generate a facial image which appears to be speaking, e.g., a "talking head."

Such an audio-visual presentation is useful in speech reading and learning applications where the posture of the mouth is important. Other applications can include electronic voice mail, animation, audio visual presentations, web based agents seeking and retrieving audio data, and interactive kiosks, such as automated teller machines. In these applications, the facial image facilitates the comprehensibility of the audible speech.

An important problem when time aligning the audio and visual signals is to make the audio-visual speech realistic. Creating a realistic appearance requires that the speech be accurately synchronized to the dynamically generated images. Moreover, a realistic rendering should distinctly reproduce, to the finest level of detail, every facial gesture which is associated with every portion of continuous natural speech.

One conventional synchronization method uses a "frame-by-frame" technique. The speech signal is analyzed and aligned to a timed sequence of image frames. This technique however lacks the ability to resynchronize in real time to perform what is called "adaptive synchronization." As a result, unanticipated real time events can annoyingly cause the synchronization to be lost.

In another technique, the dynamic images of a "talking head" are adaptively synchronized to a speech signal, see U.S. Patent 5,657, 426 from U.S.S.N. 08/258,145, "Method and Apparatus for Producing Audio-Visual Synthetic Speech" filed by Waters et al, filed on June 10, 1994. There, a speech synthesizer generates fundamental speech units called phonemes which can be converted to an audio signal. The phonemes can be translated to their visual complements called visemes, for example mouth postures. The result is a sequence of facial gestures approximating the gestures of speech.

Although the above prior technique allows a close synchronization between the audio and visual signals, there are still certain limitations and setbacks. The visual images are driven by input text, and not human speech. Also, the synthetic speech sounds far from natural, resulting in an audio-visual dichotomy between the fidelity of the images and the naturalness of the synthesized speech.

In the prior art, some techniques are known for synchronizing natural speech to facial images. In one technique, a coarse-grained volume tracking approach is used to determine speech loudness. Then, the relative opening of the mouth in the facial image can be time aligned to the audio signals. This approach, however, is very limited because mouths do not just simply open and close in an exactly known manner as speech is rendered.

An alternative technique uses a limited speech recognition system to produce broad categorizations of the speech signal at fixed intervals of time. There, a linear-prediction speech model periodically samples the audio waveform to yield an estimated power spectrum. Sub-samples of the power spectrum representing fixed-length time portions of the signal are concatenated to form a feature vector which is considered to be a "frame" of speech. The fixed length frames are typically short in duration, for example, 5, 10, or 20 microseconds (ms), and bear no relationship to the underlying acoustic-phonetic content of the signal.

Each frame is converted to a script by determining the Euclidean distance from a set of reference vectors stored in a code book. The script can then be translated to visemes. This means, for each frame, substantially independent of the surrounding frames, a "best-fit" script is identified, and this script is used to determine the corresponding visemes to display at the time represented by the frame.

The result is superior to that obtained from volume metrics, but is still quite primitive. True time-aligned acoustic-phonetic units are difficult to achieve, and this prior art technique does not detect the starting and ending of acoustic-phonetic units for each distinct and different portion of the digitized speech signal.

Therefore, it is desired to accurately synchronize visual images to a speech signal. Furthermore, it is desired that the visual images include fine grained gestures representative of every distinct portion of natural speech.

SUMMARY OF THE INVENTION

In the present invention, a computerized method is used to synchronize audio signals to computer generated visual images. A digitized speech signal acquired from an analog continuous natural speech signal is analyzed to produce a stream of time aligned acoustic-phonetic units. Acoustic-phonetic units are hypothesized for portions of the input speech signal determined to be phonetically distinct. Each acoustic-phonetic unit is associated with a starting time and an ending time of

the phonetically distinct portion of the speech signal.

The invention, in its broad form, resides in a computerized method for synchronizing audio signals to computer generated visual images, as in claim 1.

In preferred embodiments the time-aligned acoustic-phonetic units are translated to corresponding time aligned image units representative of the acoustic-phonetic units. Then, an image including the time aligned image units is displayed while synchronizing to the speech signal. The image units correspond to facial gestures producing the speech signal. The rendering of the speech signal and image can be performed in real-time as speech is generated.

In one embodiment, the acoustic-phonetic units are of variable durations, and correspond to fundamental linguistic elements. The phonetic units are derived from fixed length frames of speech processed by a pattern classifier and a phonetic recognizer using statistical trajectory models.

In another embodiment, the speech signals are acquired by a first client computer system, and the speech signal and the image are rendered in a second client computer system by communicating phonetic and audio records. Each phonetic record includes an identity of a particular acoustic-phonetic unit, and the starting and ending time of the acoustic phonetic unit.

BRIEF DESCRIPTION OF THE DRAWINGS

A more detailed understanding of the invention may be had from the following description of preferred embodiments, given by way of example, and to be read in conjunction with the accompanying drawing, wherein:

- Figure 1 is a block diagram of a audio-visual synchronization system according to a preferred embodiment of the invention;
- Figure 2 is a block diagram of a pattern classifier and pattern recognizer sub-system of the system of Figure 1; and
- Figure 3 is a block diagram of a distributed audio-visual synchronization system.

DETAILED DESCRIPTION OF PREFERRED EMBODIMENTS

Figure 1 shows a computer implemented system 100 for synchronizing audio signals, such as human speech, to visual images, such as an animated talking head rendered on a display screen 2. In Figure 1, the analog audio signals are acquired by a microphone 110. An analog-to-digital convertor (ADC) 120 translates the audio to digital signals on lines 111 and 112.

Although the example system 100 is described in terms of human speech and facial images, it should be understood that the invention can also process other audio signals and animated images, such as barking dogs, or inanimate objects capable of producing sounds with distinctive frequency and power spectrums.

A digital speech processing (DSP) sub-system 200, described in further detail below, converts the digital speech signals to time aligned acoustic-phonetic units (A-P UNITS) 113 on line 114. The units 113, which have well defined and time aligned boundaries and transitions, are acoustic realizations of their linguistic equivalents called phonemes. A translator 130 using a dictionary 131 converts the acoustic-phonetic units 113 to time-aligned visemes 115 on line 116.

The digital audio signals on line 112 can be communicated in the form of an audio file 117, for example, a "*.wav" file. The visemes 115 and the audio file 117 are processed by a rendering sub-system 240. The rendering sub-system includes output devices: a display screen 2, and a loudspeaker 3.

Figure 2 shows the DSP 200 in greater detail. A front-end preprocessor (FEP) 210 converts the digital audio signals to a temporal sequence of vectors or overlapping observation frames 211 on line 212. The frames 211 can be in the form of feature vectors including Mel-Frequency cepstral coefficients (MFCC). The coefficients are derived from short-time Fourier transforms of the digital signals. The MFCC representation is described by P. Mermelstein and S. Davies in <u>Comparison of Parametric Representation for Monosyllabic Word Recognition in Continuously Spoken Sentences</u>, IEEE Trans ASSP, Vol. 23, No. 1, pages 67-72, February 1975.

The cepstral coefficients provide a high degree of data reduction, since the power spectrum of each of the frames is represented using relatively few parameters. Each frame parameterizes a set of acoustic features which represent a portion of the digitized audio signal at a given point in time. Each frame includes, for example, the MFCC parameters.

The frames 211 are processed by a pattern classifier and phonetic recognizer (PCPR) 220. The PCPR uses a segment based approach to speech processing. The segment based approach is called statistical trajectory modeling (STM).

According to STM, each set of acoustic models comprise "tracks" and error statistics. Tracks are defined as a trajectory or temporal evolution of dynamic acoustic attributes over segments of speech. During statistical trajectory modeling, a track is mapped onto designated segments of speech of varying duration. The designated segments can be units of speech, for example, phones, or transitions from one phone to another.

The purpose of the tracks is to accurately represent and account for the dynamic behavior of the acoustic attributes over the duration of the segments of the speech signals. The error statistics are a measure of how well a track is expected to map onto an identified unit of speech. The error statistics can be produced by correlating the difference between synthetic units of speech generated from the track with the actual units of

speech. The synthetic unit of speech can be generated by "deforming" the track to conform to the underlying acoustic unit of speech.

As shown in Figure 2, the acoustic-phonetic units are formatted as data records 230. Each record 230 includes three fields. A starting time 231, an ending time 232, and an identification 233 of the corresponding acoustic-phonetic unit. The acoustic units correspond to phonetically distinct portions of the speech signal such as phones or transitions between phones. The acoustic-phonetic units are translated to visemes and further processed by the rendering sub-system 240. The rendering system can be as described in US Patent 5,657,426 supra.

Because of the statistically stationary segments produced by the STM technique, time alignment of the acoustic-phonetic units to visemes can be extremely accurate. This is particularly true for phones in consonant classes which are not handled well, if at all, by the prior art techniques.

Although, the invention has been described with respect to the visemes being related to mouth gestures, it should be understood that other facial gestures could also be synchronized, such as the eyes, eyelids, eyebrows, forehead, ears, nose, and jaw.

In one embodiment of the invention, the system components of Figure 1 can be incorporated into a single computer system.

Figure 3 shows an alternative embodiment configured as a distributed computer system 300. The distributed system 300 can use the Internet with the World-Wide-Web (WWW, or the "web") interface 310. The system 300 includes a sender client computer 320, a receiver client computer 330, and a web server computer 340.

The sender client computer 320 includes hardware and software 321 to acquire analog audio signals, and to forward the signals digitally to another client computer, for example, the receiver client 330 using Internet and WWW standard communication protocols. Such a system is described in European Patent Application S. N. 97115923.1. The web server computer 340 includes the PCPR sub-system 200 as described above. The receiver client computer 330 includes a mail receiver sub-system enhanced with the rendering sub-system 240 of Figure 1.

During operation of the system 300, a user of the sender client 320 provides an audio message for one or more recipients. The audio message can be in the form of a ".wav" file. The message is routed via the web server computer 340 to the receiver client computer 330. The PCPR 200 of the web server 340 appends the .wav file with the appropriate time-aligned phonetic records 230. Then, the user of the receiver client can "hear" the message using the mailer 331. As the message is being played back, the rendering sub-system will provide a talking head with facial gestures substantially synchronized to the audio signal.

It should be understood that the invention can also be used to synchronize visual images to streamed audio signals in real time. For example, a web-based "chat room" can be configured to allow multiple users to concurrently participate in a conversation with multiple synchronized talking heads. The system can also allow two client computers to exchange audio messages directly with each other. The PCPR can be located in either client, or any other accessible portion of the network. The invention can also be used for low-bandwidth video conferencing using, perhaps, digital compression techniques. For secure applications, digital signals can be encrypted.

The foregoing description has been directed to specific embodiments of this invention. It will be apparent, however, that variations and modifications may be made to the described embodiments, with the attainment of all or some of the advantages. Therefore, it is the object of the appended claims to cover all such variations and modifications as come within the scope of this invention.

Claims

1. A computerized method for synchronizing audio signals to computer generated visual images;

   analyzing a speech signal to produce a stream of time aligned acoustic-phonetic units, there is one acoustic-phonetic unit for each portion of speech signal determined to be phonetically distinct, each acoustic phonetic unit having a starting time and an ending time of the phonetically distinct portion of the speech signal;
   translating each acoustic-phonetic unit to a corresponding time aligned image unit representative of the acoustic-phonetic unit; and
   displaying an image including the time aligned image units while synchronizing to the speech signal.

2. The method of claim 1 further comprising:

   converting a continuous analog natural speech signal to a digitized speech signal before analyzing the speech signal.

3. The method of claim 1 wherein the acoustic-phonetic units have variable durations.

4. The method of claim 1 wherein the acoustic-phonetic units can be interpreted as fundamental linguistic elements.

5. The method of claim 1 further comprising:

   partitioning the speech signals into a sequence of frames;
   processing the frames by a pattern classifier

and phonetic recognizer, further comprising:

applying statistical trajectory models while processing the frames.

6. The method of claim 1 wherein the visemes correspond to facial gestures.

7. The method of claim 1 further comprising:

acquiring the speech signals by a first client computer system;
rendering the speech signal and the image in a second client computer system, further comprising:

communicating phonetic records between the first and second client computer systems, each phonetic record including an identity of a particular acoustic-phonetic unit, and the starting and ending time of the acoustic phonetic unit.

8. The method of claim 7 further comprising:

formatting the speech signal in an audio data file; and
appending the phonetic records to the audio data file, further wherein, the first and second client computers are connected by a network, and further comprising:

analyzing the speech signal in a server computer system connected to the network.

9. The method of claim 1 further comprising:

performing the analyzing, translating, and displaying steps synchronously in real-time.

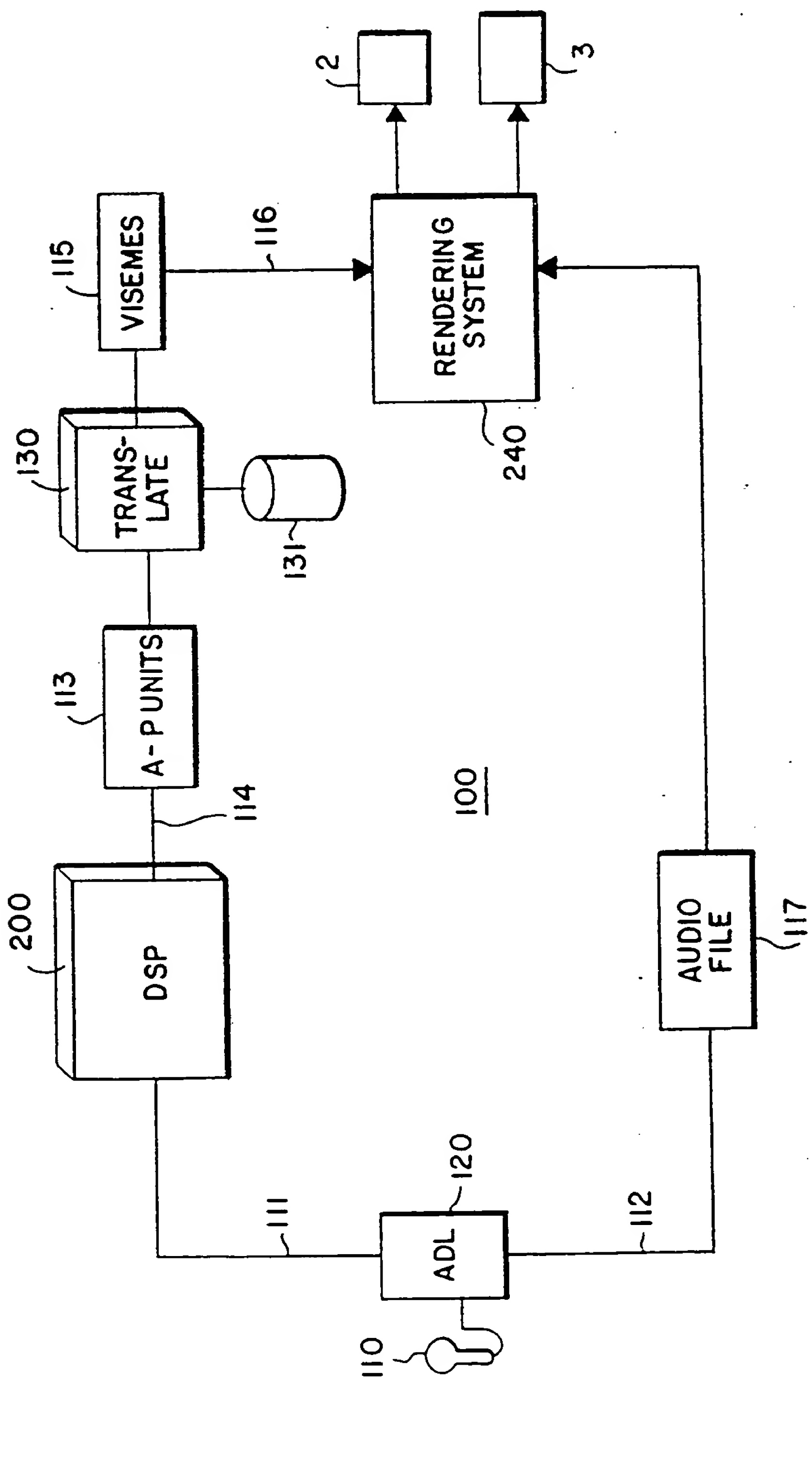
DSP
200
114
A - P UNITS
113
TRANS-
LATE
130
131
VISEMES
115
116
RENDERING
SYSTEM
240
2
3
100
AUDIO
FILE
117
ADL
120
110
111
112

FIG. 1

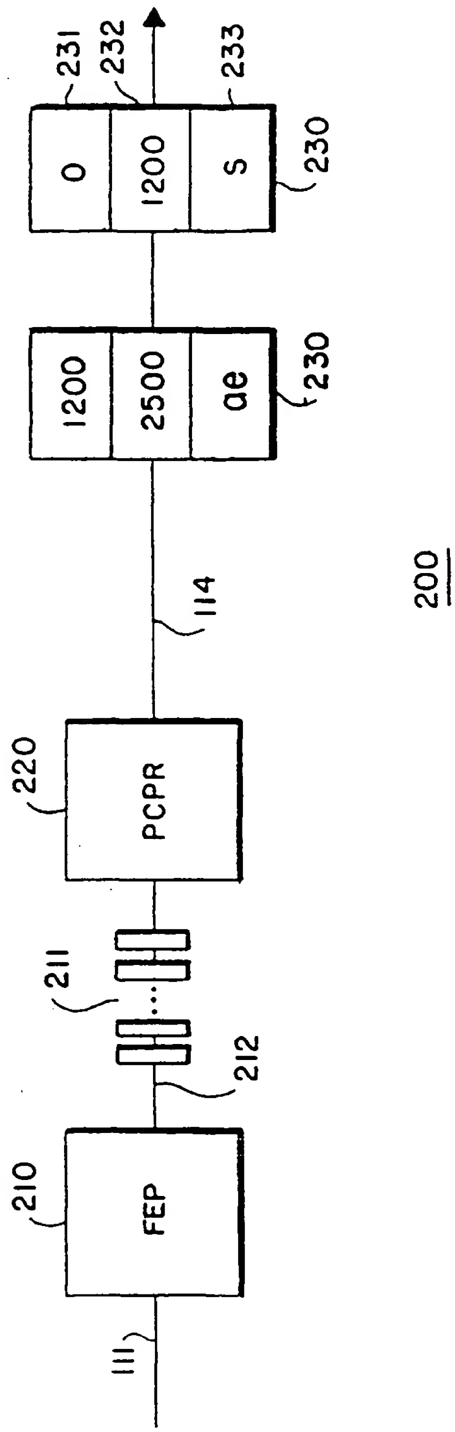
111
FEP
210
212
211
PCPR
220
114
1200
2500
ae
230
0
1200
S
230
231
232
233
200

FIG.2

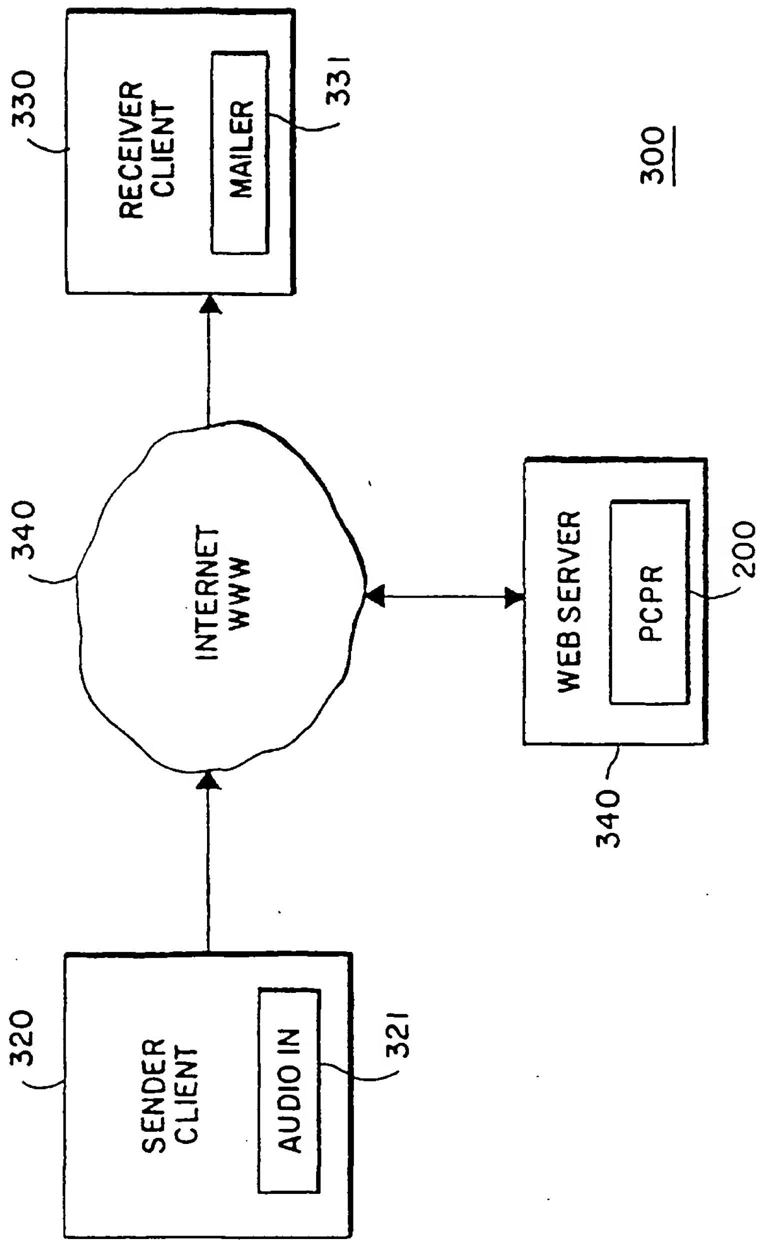
330
RECEIVER CLIENT
MAILER
331
340
INTERNET WWW
300
WEB SERVER
PCPR
200
340
320
SENDER CLIENT
AUDIO IN
321

FIG. 3